REMARKS

In accordance with the foregoing, claims 1-10 have been amended. Claims 1-10 are pending and under consideration. No new matter is presented.

REJECTION UNDER 35 U.S.C. §102:

Claims 1-7 are rejected under 35 U.S.C. 102(e) as being anticipated by Hardwick (U.S. Patent No. 6,377,916).

Claims 1-10 have been amended to clarify the present invention.

The Office Action sets forth Hardwick discloses a standard speech coder for receiving a speech signal while dividing the speech signal into spectrum information and an excited signal component and generating standing coded bit streams by performing modeling, quantizing, and coding with respect to the spectrum information and the excited signal (abstract with column 2, lines 24-38 and column 5, lines 47-65 with column 10, lines 62-65 and column 11, lines 36-46). In addition the Office Action sets forth that Hardwick discloses "a quality enhancement coder for obtaining errors between the quantized signal and the desired signal with respect to each of the spectrum information and the excited signal component, and generating coded bit streams by performing additional quantization with respect to the obtained errors(column 14, lines 25-40).

By way of review, Hardwick states "[a] set of speech model parameters then is estimated for a frame. The speech model parameters include voicing parameters dividing the frame into voiced and unvoiced regions, at least one pitch parameter representing pitch for at least the voiced regions of the frame, and spectral parameters representing spectral information for at least the voiced regions of the frame. The speech model parameters are quantized to produce parameter bits." However, Hardwick fails to disclose "a standard speech coder for dividing the speech signal into spectrum information and an excited signal component" as recited in claim 1.

Further, Hardwick states "[m]ore precise quantization of the model parameters can be achieved in many ways. However, one method which is advantageous in certain applications is to use multiple layers of quantization, where the second layer quantizes the error between the unquantized parameter and the result of the first layer, and additional layers work in a similar manner. This hierarchical approach may be applied to the quantization of the spectral magnitudes, where a second layer of quantization is applied to the spectral errors computed as a result of the first layer of quantization described above. For example, in one implementation, a second quantization layer is achieved by transforming the spectral errors with a DCT and using a vector quantizer to quantize some number of these DCT coefficients. A typical approach is to use a gain quantizer for the first coefficient plus split vector quantization of the subsequent coefficients"(col. 14, lines 25-40). Hardwick fails to disclose "a quality enhancement coder for

obtaining errors between the quantized signal and the unquantized signal with respect to the excited signal component, and generating coded bit streams by performing additional quantization with respect to the obtained errors" as recited in claim 1.

As such, it is respectfully submitted that Hardwick does not disclose the invention recited in claim 1.

Regarding claim 2, the outstanding Office Action sets forth that Hardwick discloses a transmitter wherein the quality enhancement coder quantizes each of the errors by using additional bits to perform multi-stage quantization (additional bits; column 14, lines 41-55 and column 18, lines 42-64).

Regarding claim 3, the Office Action sets forth that Hardwick discloses a transmitter wherein the quality enhancement coder uses a vector quantization method for additional quantization (vector quantization; column 14, lines 41-55).

Claim 3 has been amend to recite "The transmitter according to claim 1, wherein the quality enhancement coder uses an algebraic codebook for additional quantization."

Accordingly, it is respectfully submitted that Hardwick does not disclose the invention recited in claim 3.

In addition, claims 4 and 5 are deemed patentable due at least their depending from claim 1, as well as their additional recitations therein.

Regarding claim 6, the Office Action sets forth that Hardwick discloses "an excited signal error quantization block for receiving an unquantized excited signal (unquantized) and a quantized excited signal (quantized) from the standard speech coder and performing a quantization procedure with respect to errors of the two excited signals (column 13, line 13- column 14, line 67 with column 18, lines 29-64)."

By way of review, claim 6 recites "an excited signal error quantization block for receiving an unquantized excited signal and a quantized excited signal from the standard speech coder and performing a quantization procedure with respect to errors of the two excited signals" (emphasis added). Hardwick fails to disclose the recited invention in claim 6.

As such, it is respectfully submitted that Hardwick does not disclose the invention recited in claim 6.

In addition, claim 7 is deemed patentable due at least to its depending from claim 6, as well as for the additional recitations therein.

REJECTION UNDER 35 U.S.C. §103:

Claims 8-10 are rejected under 35 U.S.C. 103(a) as being unpatentable over Maeda (U.S. Patent No. 6,658,378) in view of Oishi (U.S. Patent No. 7,047,186).

The Office Action acknowledges that Maeda does not teach a demultiplexing block and a standard speech decoder. However the Office Action sets forth that Maeda discloses "a quality enhancement decoder for receiving a additional LSP index (LSP index) and the additional excited index and generating error components (residual) of the spectrum information and the excited signal by performing a dequantization procedure with respect to the additional LSP index and the additional excited signal index (column 24, lines 15-64)."

By way of review, claim 8 recites "a demultiplexing block for demultiplexing bit streams of the speech signal to generate an LSP (Line Spectrum Pairs) index, an excited signal index and an additional excited signal index to compensate the error of an excited signal component of the speech signal."

However, Maeda merely discloses "The index as an envelope quantization output from the terminal 203 is sent to a vector dequantizer 212 for vector quantization to find a spectral envelope of the LPC residuals which are sent to a voiced sound synthesis unit 211. The voiced sound synthesis unit 211 synthesizes LPC residuals of the voiced sound portion by sinusoidal synthesis and is fed with the pitch and voiced/unvoiced decision output from terminals 204, 205. The LPC residuals of the voiced sound from the voiced sound synthesis unit 211 are sent to an LPC synthesis filter 214. The index of the UV data from the terminal 207 is sent to the unvoiced sound synthesis unit 220 where the noise codebook is referred to in order to take out the LPC residuals as the excitation vector of the unvoiced portion. These LPC residuals are also sent to the LPC synthesis filter 214 where the LPC residuals of the voiced portion and those of the unvoiced portion are independently processed with LPC synthesis. Alternatively, the LPC synthesis may be performed on the sum of the LPC residuals of the voiced portion and those of the unvoiced portion. The LSP index from the terminal 202 is sent to an LPC parameter reproducing unit 213 to take out .alpha.-parameters of the LPC which are sent to the LPC synthesis filter 214. The speech signals, obtained on LPC synthesis by the LPC synthesis filter 214, are taken out at an output terminal 201"(col. 24, lines 14-37).

Hence, Maeda fails to disclose "an excited signal index and an additional excited signal index to compensate the error of an excited signal component of the speech signal" as recited in claim 8.

Furthermore, the Office Action sets forth that Oishi discloses "demultiplex block for receiving bit streams of a speech signal and demultiplexing the bit streams (bit-stream) of the speech signal to generate an LSP index and an additional LSP index (LSP) to compensate the error of spectrum information of the speech signal , and an excited signal index and an additional excited signal index to compensate the error of an excited signal component of the speech signal(column 5, line 65- column 6, line 13 and column 8, lines 10-12)

By way of review, Oishi discusses "the input unit 1 receives voice signal (narrow band and wide band voice signals) which are so-called a bit-stream and coded by a voice coding apparatus (not illustrated), and inputs the received signals to the de-multiplexer 2. the bit-stream includes indexes respectively corresponding to an LSP (Line Spectrum Pair), a gain, an adaptive code vector and a pulse signal. The de-multiplexer 2 divides the bit stream input from the input unit 1 into indexes respectively corresponding to an LSP, a gain, an adaptive code vector and a pulse signal. Specifically, the de-multiplexer 2 provides the LSP decoding unit 3 with the LSP index, the band determiner 4 with the gain index, the adaptive-code book decoding unit 5 with the adaptive code vector index and the pulse-code-book decoding unit 6 with the pulse signal index. The LSP decoding unit 3 generates LSPs by decoding the provided LSP index, and outputs the generated LSP to the band determiners 4, 7 and 13."(col. 5 line 65 to col. 6, line 15).

As such, Oishi fails to disclose "a demultiplexing block for demultiplexing bit streams of the speech signal to generate an LSP(Line Spectrum Pair) index, an excited signal index and an additional excited signal index to compensate the error of an excited signal component of the speech signal" as recited in claim 8.

Accordingly, it is respectfully submitted that the combination of Maeda and Oishi does not teach or suggest the invention recited in claim 8.

Regarding claim 9, the Office Action sets forth that Maeda discloses "an excited signal dequantization block for receiving the excited signal index and restoring the excited signal by performing a dequantization procedure with respect to the excited signal index (column 24, lines 15-64)

By way of view, Maeda discloses "[t]he index of the UV data from the terminal 207 is sent to the unvoiced sound synthesis unit 220 where the noise codebook is referred to in order to take out the LPC residuals as the excitation vector of the unvoiced portion. These LPC residuals are also sent to the LPC synthesis filter 214 where the LPC residuals of the voiced portion and those of the unvoiced portion are independently processed with LPC synthesis. Alternatively, the LPC synthesis may be performed on the sum of the LPC residuals of the voiced portion and those of the unvoiced portion. The LSP index from the terminal 202 is sent to an LPC parameter

reproducing unit 213 to take out α parameters of the LPC which are sent to the LPC synthesis filter 214. The speech signals, obtained on LPC synthesis by the LPC synthesis filter 214, are taken out at an output terminal 201" (col. 24, lines 24-37)

As explained above and FIG. 24 , Maeda fails to disclose "an excited signal dequantization block for receiving the excited signal index from the demultiplexed bit streams of the speech signal and restoring the excited signal by performing a dequantization procedure with respect to the excited signal index" as recited in claim 9.

Accordingly, it is respectfully submitted that the combination of Maeda and Oishi does not teach or suggest the invention recited in claim 9.

In addition, claim 10 also deemed patentable due at least to its depending from claim 8, as well as the additional recitations therein.

CONCLUSION:

There being no further outstanding objections or rejections, it is submitted that the application is in condition for allowance. An early action to that effect is courteously solicited.

Finally, if there are any formal matters remaining after this response, the Examiner is requested to telephone the undersigned to attend to these matters.

If there are any additional fees associated with filing of this Amendment, please charge the same to our Deposit Account No. 19-3935.

Respectfully submitted,

STAAS & HALSEY LLP

Date: 07/30/2007

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